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APPLICATION BULLETIN
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LOW SAMPLING RATE OPERATION FOR BURR-BROWN SoundPLUS™ AUDIO DATA CONVERTERS AND CODECS

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PURPOSE

This application bulletin describes the operation and performance of Burr-Brown's PCM digital audio products when using low sampling frequencies.

INTRODUCTION

For most consumer audio applications, customers using Burr-Brown's PCM digital audio products will operate at a standard audio sampling frequency, such as 44.1kHz, 48kHz, or 96kHz. However, there are applications where lower sample rates are either desirable or required. These applications include computer audio, telephones, intercom systems, speech processing, video teleconferencing, and modems. For these applications, sample rates from 8kHz to 22.05kHz are very common.

Although Burr-Brown concentrates its PCM design efforts to obtain specified performance using sampling frequencies in the 32kHz to 96kHz range, most PCM products can support sampling rates as low as 4kHz while achieving near typical dynamic performance. The following sections will examine the low sampling frequency operation and performance of Burr-Brown's PCM products, with emphasis on understanding the operational and performance limitations, as well as circuit design considerations for these applications.

GENERAL CONSIDERATIONS FOR DELTA-SIGMA DATA CONVERTERS

For delta-sigma data converters, theoretical dynamic performance is given by the performance of the digital filter, modulator, and output amplifier sections. Delta-sigma data converters use high oversampling rates, such as $64f_S$, or 64 times the desired sample rate. This oversampling in the digital filter and modulator section automatically tracks with changes in sample frequency, f_S. Overall performance of the digital filter and modulator remains relatively constant over the range of usable sampling frequencies.

Circuitry external to the data converter also impacts the overall dynamic performance. These circuits include low pass filters used for digital-to-analog (D/A) post filtering and analog-to-digital (A/D) anti-aliasing filtering. These filters are required in order to limit the output and input signal bandwidth.

DYNAMIC LOGIC AND ITS EFFECT ON MINIMUM SAMPLE RATE FOR DELTA-SIGMA CONVERTERS

Delta-sigma data converters utilize dynamic logic circuits, especially in their oversampling digital filter sections. Figures 1 and 2 show the equivalent circuit and transistor level circuit diagram of a dynamic logic register. The dynamic

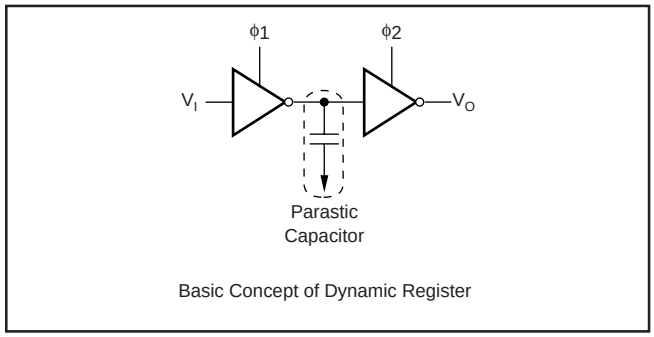

FIGURE 1. Simple Equivalent Circuit for Dynamic Register.

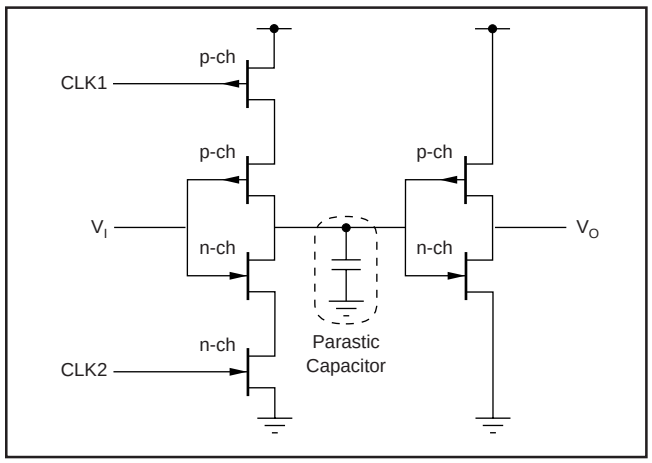

FIGURE 2. Transistor Level Circuit Diagram of Dynamic Register.

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register relies upon the charge stored on a parasitic capacitance and the clocks which are used to transfer the stored state from the input to the output of the register. The clock rate is directly related to the sampling rate of the converter.

The parasitic capacitance value associated with the dynamic register is very small. Additionally, there are leakage currents associated with the MOS devices in the register, even when they are turned off. Given the effects of these leakage currents, charge on the capacitor can only be held for a small amount of time. Since the clocks used for the dynamic register track with the sampling frequency, lower sampling rates translate into a longer storage period. This in turn leads to a larger amount of stored charge being dissipated due to leakage current. If the sampling rate becomes too low, so much of the charge is dissipated that it can result in data errors and non-functional behavior.

Table I shows the minimum sample rate for Burr-Brown's delta-sigma data converters and CODECs. Most can operate with a sampling frequency as low as 4kHz, due to the limitations of dynamic logic described in the previous paragraphs. The PCM1723 and PCM1727 can operate at a minimum sample frequency of 16kHz, due to the limitations of the on-chip Phase Locked Loop (PLL) circuitry used to generate audio system clocks.

	SAMPLING FREQUENCY (kHz)[1,2]	
PRODUCT	MIN	MAX
PCM1704	n/a	768
PCM1716	4	96
PCM1717	4	48
PCM1718	4	48
PCM1719	4	48
PCM1720	4	96
PCM1723[3]	16	96
PCM1725	4	96
PCM1727[3]	16	96
PCM1728	4	96
PCM1733	4	96
PCM1800	4	48
PCM3000	4	48
PCM3001	4	48
PCM3002	4	48
PCM3003	4	48
PCM3006	4	48

NOTES: (1) AC specifications are not guaranteed over the entire frequency range. (2) Maximum sampling frequency is limited by internal timing requirements. (3) Minimum sampling frequency is limited by on-chip Phase Locked Loop.

Table I. Minimum and Maximum Sampling Frequencies for PCM Products.

MINIMUM SAMPLE RATE FOR SIGN MAGNITUDE D/A CONVERTERS

Sign magnitude D/A converters, like the PCM1704, do not employ delta-sigma conversion techniques and are similar to conventional R-2R ladder DACs. These converters do not have a low sampling frequency limit since they use a simple latched serial interface and employ no digital filtering or oversampling modulator circuitry. These converters can be operated at rates down to DC. For more information, refer to the "Stopped Clock Operation" sections of the PCM1702 and PCM1704 data sheets.

DIGITAL AND ANALOG FILTERING AT LOW SAMPLING FREQUENCIES

For delta-sigma data converters, the frequency response of the on-chip digital interpolation and decimation filters is dictated by the sampling rate, f_S, where passband and stopband frequencies track with changes in f_S. Since the filter response tracks with sampling frequency, the digital filter maintains its characteristics at lower sampling frequencies. As an example, Figure 3 shows the overall frequency response of the interpolation filter used in the PCM1716 stereo DAC. Using this plot, the frequency response of the digital interpolation filter can be determined for any sampling frequency over the 4kHz to 96kHz range. However, the RC continuous time analog filter used in the output stage of D/A converters, as well as the input anti-alias filter used at the input of A/D converters, have a fixed frequency response that does not track with sampling frequency. Figures 4 shows the overall and passband frequency response of the internal output filter for our D/A converters. Figure 5 shows the overall and passband frequency response of the internal anti-aliasing filter for our A/D converters.

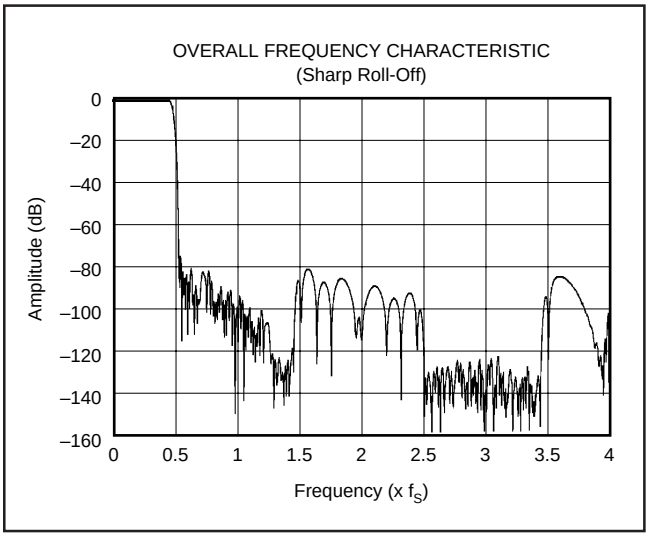


FIGURE 3. PCM1716 Digital Filter Response.

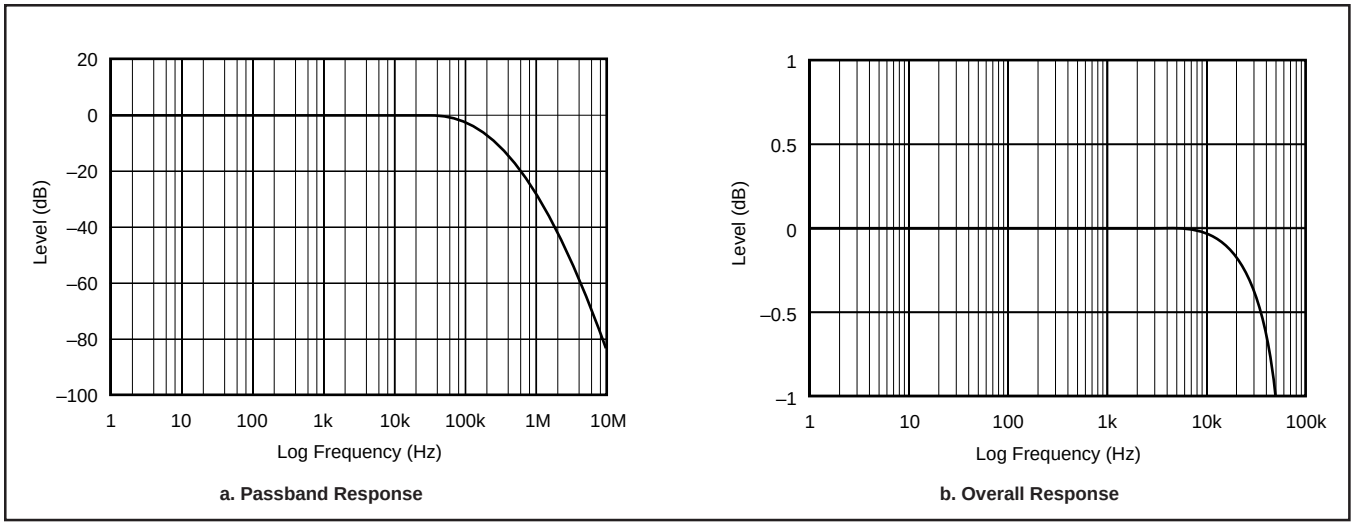


FIGURE 4. Frequency Response of DAC Output Filter.

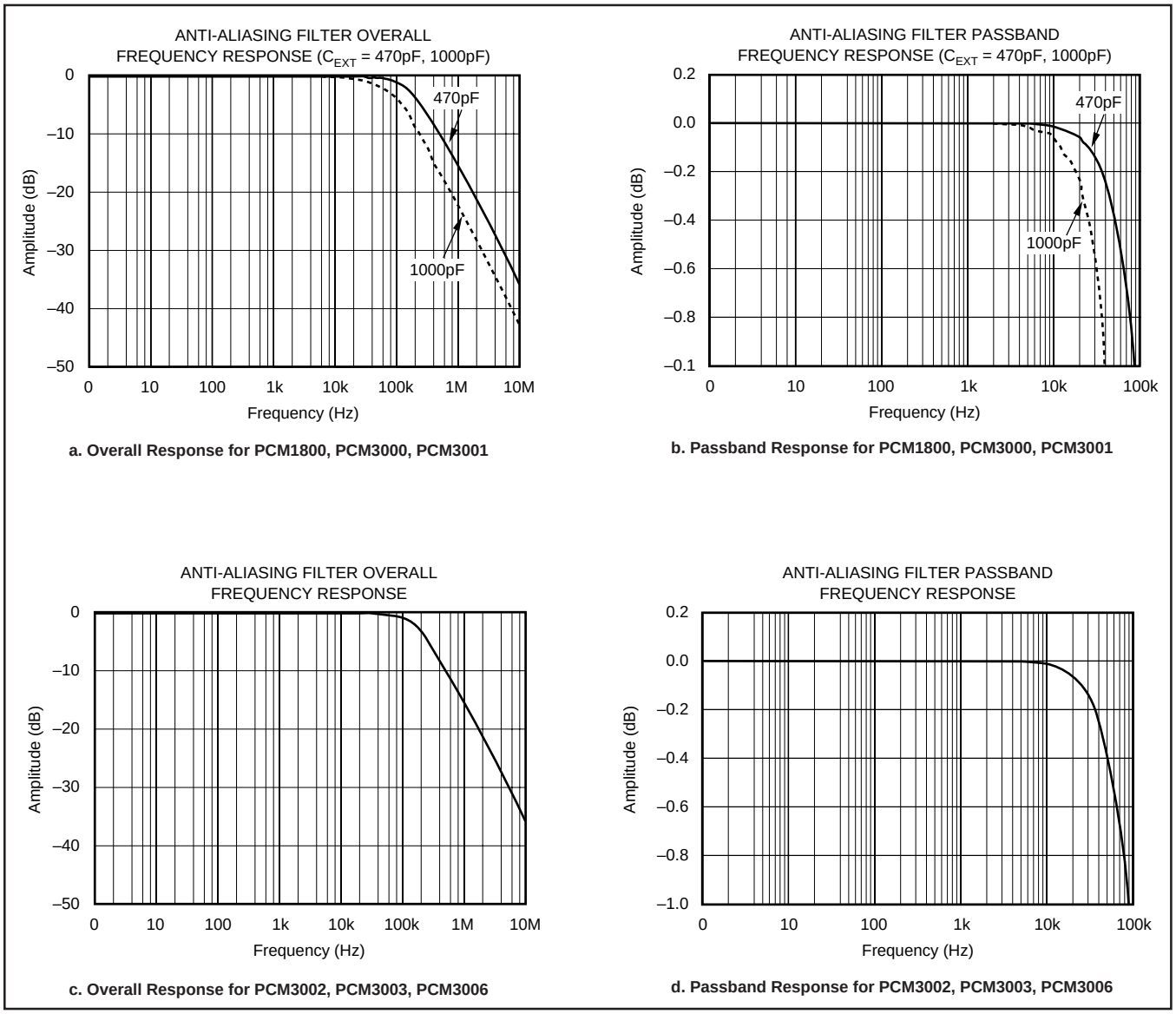
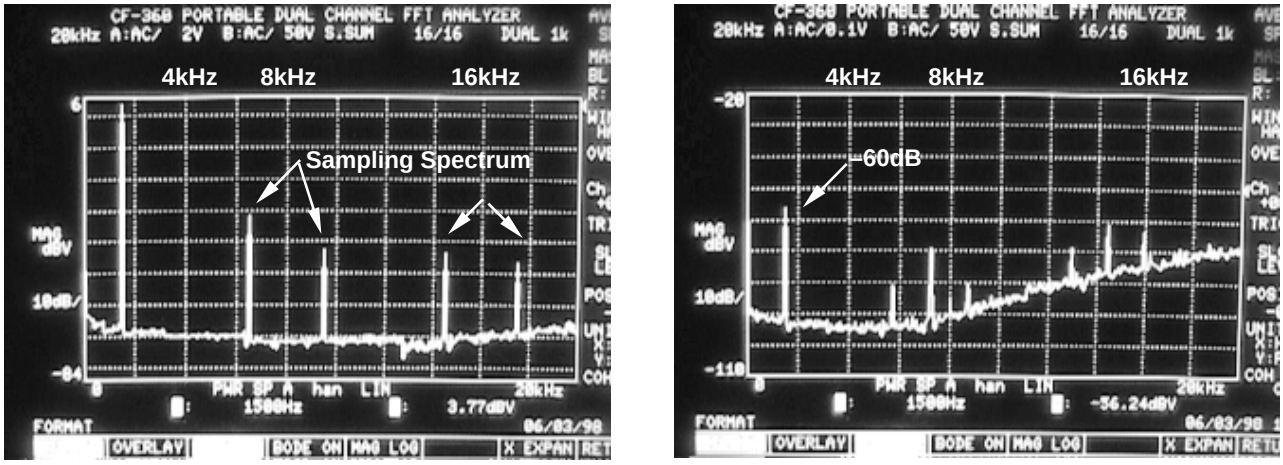


FIGURE 5. Frequency Response of ADC Anti-Aliasing Filter.

Using D/A converters at low sampling rates (f_S = 4kHz to 16kHz), the noise spectrum above the Nyquist frequency ($f_S/2$) is audible, and is not affected by the analog filter since the spectrum is well within its passband. Figure 6 shows an example where the PCM1717 stereo DAC is being operated with f_S = 8kHz. The delta-sigma modulator noise and sampling spectrum are clearly visible in these photos. External low pass filtering is required to attenuate the frequency spectrum above the Nyquist frequency.

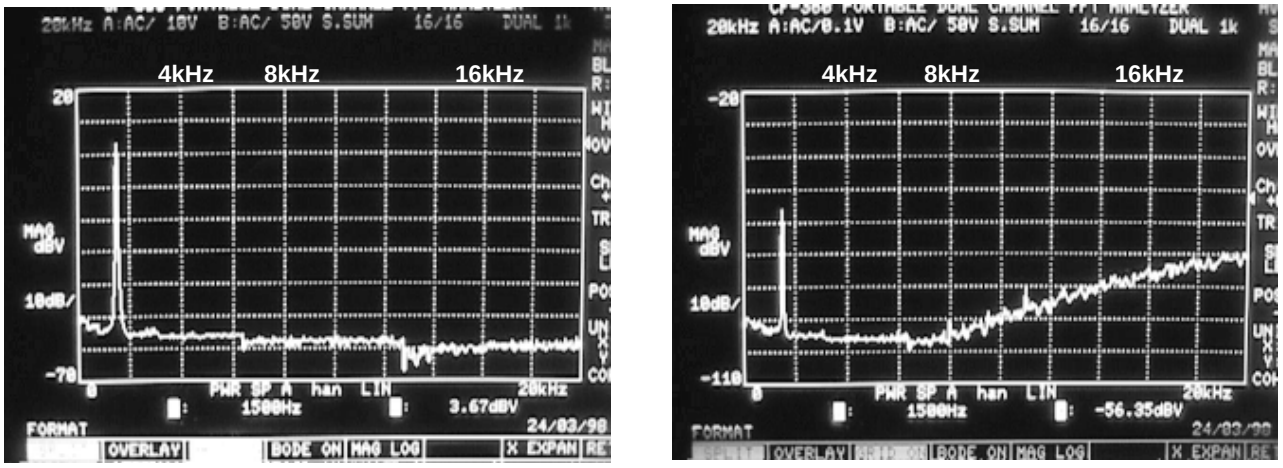
Figure 7 shows another example where the PCM1716 is operated with f_S = 8kHz. The sampling spectrum is not present due to the improved stopband attenuation of the PCM1716's digital interpolation filter. However, the delta-sigma modulator noise is clearly present. Once again, an external lowpass filter is necessary to attenuate the out-of-band noise spectrum to an acceptable level. The amount of filtering required will be determined by the out-of-band noise requirements for a given application.



a. Full-Scale Output

b. –60dB Output

FIGURE 6. PCM1717 Output Spectrum with f_S = 8kHz.



a. Full-Scale Output

b. –60dB Output

FIGURE 7. PCM1716 Output Spectrum with f_S = 8kHz.

Figure 8 shows a simple 2nd-order multiple feedback (MFB) active filter design that can be used to build a DAC post filter. Multiple 2nd-order sections, along with simple 1st-order RC sections, may be cascaded to create higher order filters as needed in order to obtain the desired out-of-band noise performance for a given application. However, care should be taken when designing higher order filters to preserve phase relationships critical in certain audio applications, such as surround sound processing. Refer to Burr-Brown application bulletin AB-034 for more details regarding multiple feedback (MFB) and Sallen-Key active filter design.

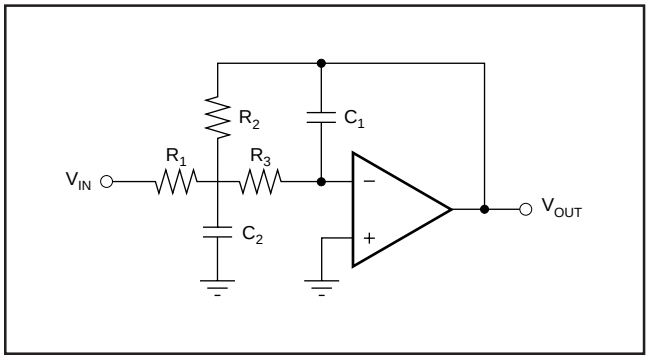


FIGURE 8. Second-Order Lowpass Filter (multipule feedback type).

For delta-sigma A/D converters, a simple single-pole RC filter is used as an anti-aliasing filter, due to the high oversampling rate used by the modulator circuit. For the PCM1800, PCM3000, and PCM3001, this filter is comprised of two on-chip resistors and an external capacitor, as shown in Figure 9. When operating with low sample rates, the value of the external capacitor should be adjusted in order to safely bandlimit the input signal. The formulas for calculating the cut-off frequency of the filter and determining the necessary external capacitor value for a given cut-off frequency are also shown in Figure 9. The cut-off frequency itself is dependent upon the desired input bandwidth for a given application.

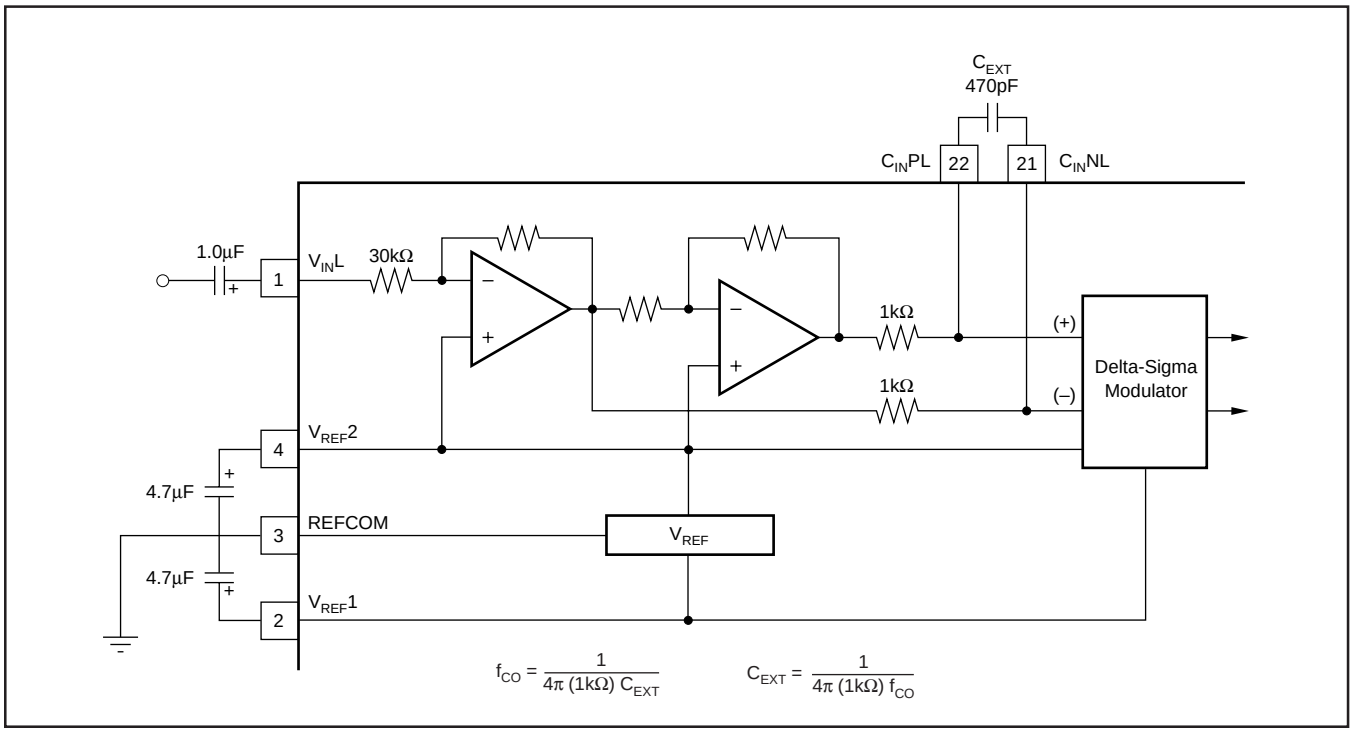


FIGURE 9. Input Circuit and Filter for PCM1800, PCM3000, PCM3001.

For the PCM3002, PCM3003, and PCM3006, the anti-alias filter is entirely on-chip, therefore, its component values cannot be altered by the user. An external RC filter is required to band limit the input for lower sampling rates. Figure 10 shows an example of adding a simple RC lowpass filter to the input of these CODECs.

DYNAMIC PERFORMANCE AT LOW SAMPLING FREQUENCIES

In order to properly measure dynamic performance at low sampling frequencies, the measurement bandwidth must be limited to the Nyquist bandwidth ($f_S/2$). For f_S = 8kHz, the measurement bandwidth must be limited to 4kHz. With this bandlimit, dynamic performance will be the same as that measured for standard sampling frequencies, such as data sheet specifications given with f_S = 44.1kHz or 48kHz.

Table II shows measurement results of THD+N for several PCM audio D/As using a 3rd-order lowpass filter and a "brickwall" 4kHz lowpass filter. The results clearly show that dynamic performance at low sampling frequency can match that of standard audio sampling rates, given that band limiting to the Nyquist frequency is provided.

For practical applications, the choice of lowpass filter order is dependent upon the actual dynamic performance (THD+N, dynamic range, SNR) and out-of-band noise level required by the system specifications.

SUMMARY

This application bulletin shows that the application spectrum of Burr-Brown's PCM digital audio products can be extended to systems requiring sampling frequencies as low as 4kHz. Typical dynamic performance can be obtained at lower sampling rates provided that sufficient low pass filtering is used by the system designer. Filter requirements should be evaluated using Burr-Brown's DEM-PCMxxxx demonstration boards and audio measurement equipment prior to finalizing the filter design so that the required dynamic performance is achieved in the final product.

PRODUCT	FILTER	MEASURED THD+N[1,2]
PCM1717	3rd-Order[3] 4kHz BW[4]	0.0265% 0.0028%
PCM1716 (16-bit data)	3rd-Order 4kHz BW	0.005% 0.0021%
PCM1716 (20-bit data)	3rd-Order 4kHz BW	0.005% 0.0019%

NOTES: (1) f_S = 8kHz, f_{IN} = 1kHz at 0dB level. (2) Measurements taken with Shibasoku 725 Distortion Analyzer (400Hz HPF, 30 kHz LPF, 20kHz BW limit). (3) 3rd-order refers to 3rd-order lowpass filter with 4kHz passband. (4) 4kHz BW refers to a "brickwall" lowpass filter with 4kHz passband.

TABLE II. Measured Performance with f_S = 8kHz.

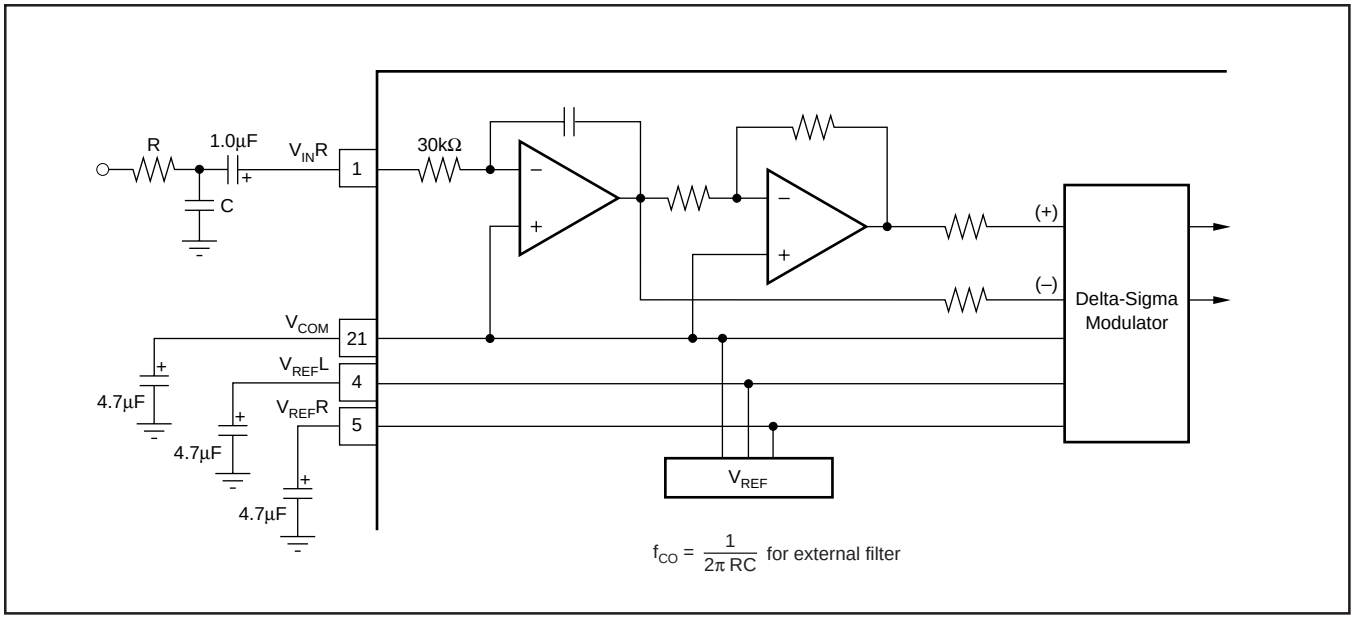


FIGURE 10. Input Circuit and Filter for PCM3002, PCM3003, PCM3006.

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